IN THE ABSTRACT

Please replace the Abstract in its entirety with the Abstract attached hereto.

IN THE CLAIMS

Please amend the claims as follows.

1. (Amended) A method for improving the quality of an audio transmission in which audio data [containing] <u>including</u> samples [(A1, ..., A8)] of an audio signal are asynchronously transmitted in data packets from a transmitting communication system [(PBX1)] via a packet-oriented communication network [(LAN)]to a receiving communication system [(PBX2)] and an information item relating to the transmission of data packets is detected, [characterized in that] <u>wherein</u> the audio data are converted [in the sense of an alteration of] <u>such that</u> their sampling rate <u>is altered</u> by means of digital filtering, the sampling rate being altered <u>based</u> [in dependence] on the detected information item, in such a manner that due to the altered sampling rate, a quality of service of the audio transmission is optimized with regard to a current transmission situation indicated by the detected information item.

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2. (Amended) [The] <u>A</u> method for improving the quality of an audio transmission in which audio data [containing] <u>including</u> samples of an audio signal are asynchronously transmitted in data packets from a transmitting communication system [(PBX1)] via a packet-oriented communication network [(LAN)] to a receiving communication system [(PBX2)] and an information item relating to the transmission of data packets is detected, [characterized in that] <u>wherein</u> the audio data are digitally converted [in the sense of a modification of] <u>such that</u> the duration [(L1+2*L2+L3)] of an audio signal represented by the audio data <u>is modified while</u>

[whilst largely] retaining a pitch of the audio signal, the duration [(L1+2*L2+L3)] being modified [in dependence] based on the detected information item, in such a manner that due to the modified duration [(L1+L2+L3)], a quality of service of the audio transmission is optimized with regard to a current transmission situation indicated by the detected information item.

Cancel claim 3.

4. (Amended) The method as claimed in claim 1, wherein [one of the preceding claims, characterized in that] the audio data to be transmitted are converted by the transmitting communication system [(PBX1)] and a conversion message about the conversion is transmitted from the transmitting communication system to the receiving communication system [(PBX2)].

5. (Amended) The method as claimed in claim 4, wherein [characterized in that] the transmitted audio data are reconverted by the receiving communication system [(PBX2)], the change in the audio data taking place in the reconversion being determined by means of the conversion message transmitted.

6. (Amended) The method as claimed in claim 1, wherein [one of the preceding claims, characterized in that] the transmission of the data packets is monitored by the receiving communication system [(PBX2)] and an information item relating to this transmission is transmitted to the transmitting communication system [(PBX1)] and [that] the audio data are converted by the transmitting communication system [(PBX1) in dependence] based on the information item transmitted.

7. (Amended) The method as claimed in claim 6, wherein [characterized in that] the information item transmitted specifies a data packet loss rate and [that], if the data packet loss

rate rises, the audio data are converted by the transmitting communication system [(PBX1)] in such a manner that the audio data rate is reduced.

8. (Amended) The method as claimed in claim 1, wherein [one of the preceding claims, characterized in that] a detected incorrect adaptation of the data rate of the received audio data is at least partially compensated by the receiving communication system [(PBX2)] by means of a conversion of the received audio data.

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9. (Amended) The method as claimed in claim 1, wherein [one of the preceding claims, characterized in that] the received audio data are converted after having been read out of an input buffer provided for compensating data packet delay variations, in which the read-out speed of the input buffer is controlled by a change in the audio data rate due to the conversion.

10. (Amended) The method as claimed in claim 1, wherein [one of the preceding claims, characterized in that,] in the case of [the] a loss of [a] one of the data packets, the audio data [contained] included in [a] the data packet preceding and/or following the lost data packet are converted by the receiving communication system [(PBX2) in the sense of an extension of] such that the duration of an audio signal represented by the audio data is extended, in such a manner that the extension of the duration at least shortens a gap in the audio signal due to the lost data packet.

11. (Amended) A communication system [(PBX1, PBX2)] for transmitting and/or receiving audio data [containing] including samples [(A1, ..., A8)] of an audio signal via a packet-oriented communication network [(LAN)], comprising:

a monitoring [means (W1, W2)] unit for detecting an information item relating to the transmission of data packets [containing] including audio data[,] ;

[characterized by] a digital sampling rate conversion device [(AU1A, AU1B, AU2A, AU2B)] for converting the audio data [in the sense of] by altering their sampling rate; and

a control [means (ST1, ST2)] unit for controlling the sampling rate alteration [in dependence] based on the information item detected.

12. (Amended) [The] A communication system [(PBX1, PBX2)] for transmitting and/or receiving audio data [containing] including samples of an audio signal via a packet-oriented communication network [(LAN)], comprising:

a monitoring [means (W1, W2)] unit for detecting an information item relating to the transmission of data packets containing audio data[,] ;

[characterized by] a digital timescale conversion device [(ZU1A, ZU1B, ZU2A, ZU2B)] for converting the audio data [in the sense of] by changing the duration [(L1+2*L2+L3)] of an audio signal represented by the audio data [whilst largely] while retaining a pitch of the audio signal[,] ; and

a control [means (ST1, ST2)] unit for controlling the change in duration [in dependence] based on the information item detected.

Cancel claim 13.

14. (Amended) The communication system as claimed in claim 11, [or 13, characterized in that] wherein the digital sampling rate conversion device [(AU1A, AU1B, AU2A, AU2B)] exhibits a digital filter chip for converting the audio data.

15. (Amended) The communication system as claimed in claim 12, [or 13, characterized in that] wherein the digital timescale conversion device [(L1+2*L2+L3)] exhibits a digital signal processor for converting the audio data.